Metaswitch VP2510 Integrated Softswitch: Connecting Cisco Unified Communications Manager 8.6 via the Cisco Unified Border Element 8.7 using SIP

Note: Testing was conducted at TekVizion labs.
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Introduction

Service Providers today, such as Metaswitch, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and a centralized IP to TDM gateway to provide on-net and off-net services. Metaswitch is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity via either Analog or T1 lines. A demarcation device between these services and customer owned services is recommended. The Cisco Unified Border Element provides demarcation, security, interworking and session management services.

- This application note describes how to configure a Cisco Unified Communications Manager (Cisco UCM) 8.6 with a Cisco Unified Border Element (Cisco UBE) 8.7 for connectivity to Metaswitch VP2510 Integrated Softswitch. The deployment model covered in this application note is CPE to PSTN. This document does not address 911 emergency outbound calls. For 911 feature service details contact Metaswitch, directly.

- Testing was performed in accordance to Cisco’s SIP Trunk Test Plan and all features were verified. Key features verified are:
  - CPE outbound to SP Offnet gateway (PSTN) (G.729 is offered first)
  - SP offnet gateway (PSTN) inbound to CPE (G.729 offered first)
  - CPE to CPE (place call out to the SP network and back) (G.729 is offered first)
  - CPE Calling number privacy
  - CPE Telephone Number Support – digit translations
  - CPE Calling Name Delivery
  - CPE offnet Call Conference
  - CPE Intra-Site Call Conference
  - CPE Intra-Site Attended Call Transfer
  - CPE Intra-Site Unattended Call Transfer
  - CPE Intra-Site Blind Call Transfer
  - CPE Call Hold and Resume (call hold is always done on the IP PBX side)
  - CPE Voice Mail
  - SP Voice Mail
  - CPE Find Me (CFU)
  - CPE T.38 FAX SG3 (G.729 is offered first)
  - Simultaneous Calls
  - CPE Auto Attendant
  - CPE to PSTN Offnet gateway international call
  - CPE G.711 FAX SG3
  - CPE Find Me (Call Forward On Busy)
  - CPE Find Me (Call Forward Don’t Answer)
  - Codec mid-call re-negotiation (to be tested without transcoder)
• PRACK with SDP (early-media cut-through with DTMF (RFC2833) navigation before 200OK)) - call 800-864-8331 - United Airlines

• The Cisco Unified Border Element configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between MetaSwitch SIP network and Cisco Unified Communications. The configuration described in this document details the important commands to have enabled for interoperability to be successful and care must be taken, by the network administrator deploying Cisco UBE, to ensure these commands are set per each dial-peer requiring to interoperate to MetaSwitch SIP network.

• This application note does not cover the use of calling search spaces (CSS) or partitions on Cisco Unified Communications Manager. To understand and learn how to apply CSS and partitions refer to the cisco.com link below:

Network Topology

Figure 1 Basic Test Environment

System Components

Hardware Components

<table>
<thead>
<tr>
<th>Component</th>
<th>Qty</th>
<th>Hardware</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Manager</td>
<td>1</td>
<td>MCS7835</td>
<td></td>
</tr>
<tr>
<td>CUBE</td>
<td>1</td>
<td>CISCO3945</td>
<td></td>
</tr>
<tr>
<td>PSTN GW</td>
<td>1</td>
<td>C3845</td>
<td>To handle PSTN calls</td>
</tr>
<tr>
<td>Switch</td>
<td>1</td>
<td>C6509</td>
<td>Lab Switch</td>
</tr>
<tr>
<td>Cisco Phones</td>
<td>4</td>
<td>7962,9971</td>
<td>Cisco Phones used as CPE.</td>
</tr>
<tr>
<td>Metaswitch VP2510 Integrated Softswitch</td>
<td>1</td>
<td>-</td>
<td>Third Party SIP Trunk Provider</td>
</tr>
</tbody>
</table>

Software Requirements

<table>
<thead>
<tr>
<th>Component</th>
<th>Qty</th>
<th>Software Version</th>
<th>Firmware Version</th>
</tr>
</thead>
<tbody>
<tr>
<td>Metaswitch VP2510 Integrated Softswitch</td>
<td>1</td>
<td>V7.3.00 SU56</td>
<td></td>
</tr>
<tr>
<td>CUBE (10.70.10.15)</td>
<td>1</td>
<td>c3900e-universalk9-mz.SPA.151-4.M4.bin</td>
<td></td>
</tr>
<tr>
<td>Communications Manager</td>
<td>1</td>
<td>8.6.1.20000-1</td>
<td></td>
</tr>
<tr>
<td>Local PSTN GW</td>
<td>1</td>
<td>C3845-advipservicesk9-mz, Version 12.4(24)T</td>
<td></td>
</tr>
<tr>
<td>Main Switch</td>
<td>1</td>
<td>Cisco WS-C6509 s3223-adventerprisek9_wan-mz, Version 12.2(33)</td>
<td></td>
</tr>
<tr>
<td>Cisco Phone 7960 (x2644)</td>
<td>1</td>
<td>P0S3-8-12-00</td>
<td></td>
</tr>
<tr>
<td>Cisco Phone 9971 (x2700)</td>
<td>1</td>
<td>sip9971.9-1-1SR1</td>
<td></td>
</tr>
<tr>
<td>Cisco Phone9672(x2604)</td>
<td>1</td>
<td>SCCP42.9-2-1S</td>
<td></td>
</tr>
</tbody>
</table>
Features

Features Supported

- Call from/to PSTN to/from CPE – Basic and International calls, digit translations
- Hold/Resume
- DTMF
- Call transfers – attended, blind, unattended
- Call Forwarding (CFU, CFB, CFNA)
- FAX: T.38 and G711ulaw
- Support for early media

Features Not Supported

- Connected party update
Caveats

- As connected party update is currently not supported by Metaswitch, CLID updates are not observed on call transfer scenarios.

- Metaswitch doesn't allow T.38 to T.38 or T.38 to G711 calls to traverse their UMG. Therefore the FAX calls were tested with media flowing directly from the CUBE to the PSTN gateway.

- During the call transfer scenarios, no ring back is heard on the transferring party when a transfer is initiated i.e.: When A and B are in call and B initiates a transfer to C, B does not hear a ring back when C is ringing. Even after B transfers the call in the case of an unattended transfer, A doesn’t hear the ring back.
Configuration

Configuring Cisco Unified Border Element

Show version:

CUBE# show ver
Cisco IOS Software, C3900e Software (C3900e-UNIVERSALK9-M), Version 15.1(4)M4,
RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2012 by Cisco Systems, Inc.
Compiled Tue 20-Mar-12 22:07 by prod_rel_team

ROM: System Bootstrap, Version 15.1(1r)T4, RELEASE SOFTWARE (fc1)

CUBE uptime is 19 hours, 16 minutes
System returned to ROM by reload at 18:52:52 UTC Mon Jun 25 2012
System restarted at 18:56:47 UTC Mon Jun 25 2012
System image file is "flash:c3900e-universalk9-mz.SPA.151-4.M4.bin"
Last reload type: Normal Reload
Last reload reason: Reload Command

This product contains cryptographic features and is subject to United
States and local country laws governing import, export, transfer and
use. Delivery of Cisco cryptographic products does not imply
third-party authority to import, export, distribute or use encryption.
Importers, exporters, distributors and users are responsible for
compliance with U.S. and local country laws. By using this product you
agree to comply with applicable laws and regulations. If you are unable
to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:

If you require further assistance please contact us by sending email to
export@cisco.com.

Cisco CISCO3945-CHASSIS (revision 1.0) with C3900-SPE250/K9 with 2064384K bytes of
memory.
Processor board ID FTX1541A032
4 Gigabit Ethernet interfaces
2 Voice FXS interfaces
DRAM configuration is 72 bits wide with parity enabled.
256K bytes of non-volatile configuration memory.
500472K bytes of ATA System CompactFlash 0 (Read/Write)

License Info:

License UDI:

---------------------------------------------------------------
Device# PID SN
---------------------------------------------------------------
*0 C3900-SPE250/K9 FOC15391VLH

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Technology Package License Information for Module:'c3900e'

<table>
<thead>
<tr>
<th>Technology</th>
<th>Technology-package</th>
<th>Type</th>
<th>Next reboot</th>
</tr>
</thead>
<tbody>
<tr>
<td>ipbase</td>
<td>ipbasek9</td>
<td>Permanent</td>
<td>ipbasek9</td>
</tr>
<tr>
<td>security</td>
<td>None</td>
<td>None</td>
<td>None</td>
</tr>
<tr>
<td>uc</td>
<td>uck9</td>
<td>Permanent</td>
<td>uck9</td>
</tr>
<tr>
<td>data</td>
<td>None</td>
<td>None</td>
<td>None</td>
</tr>
</tbody>
</table>

Configuration register is 0x2102

Configuration:

CUBE#sho run
Building configuration...

Current configuration : 14018 bytes

! Last configuration change at 19:28:57 UTC Mon Jun 18 2012 by administrator
! NVRAM config last updated at 19:28:59 UTC Mon Jun 18 2012 by administrator
! NVRAM config last updated at 19:28:59 UTC Mon Jun 18 2012 by administrator

version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
hostname CUBE
!
boot-start-marker
boot system flash c3900e-universalk9-mz.SPA.151-4.M4.bin
boot-end-marker
!
logging buffered 51200 warnings
enable secret 5 $1$lbTj$25nxZBh5UBjnXOrx9aRvn/
!
no aaa new-model
!
no ipv6 cef
ip source-route
!
ip cef
!
!
ip domain name yourdomain.com
ip name-server 10.85.0.232
multilink bundle-name authenticated
!
!
!
crypto pki token default removal timeout 0
crypto pki trustpoint TP-self-signed-3709846528
enrollment selfsigned
subject-name cn=IOS-Self-Signed-Certificate-3709846528
revocation-check none
rsakeypair TP-self-signed-3709846528
!
!
crypto pki certificate chain TP-self-signed-3709846528
certificate self-signed 01
3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
31312F30 20D00355 0401312D 49F532D5 3656C66 2D536967 6E65642D 43657274
6966696E 6174652D 33373039 38343635 3238301E 170D3131 31303039 30303538
34325A17 0D123030 31303030 305A3030 312F302D 06035504 03132649
4F532D5D 5656C66 2D5642D4 65627469 66966361 74652D33 37039389
34363532 3830819F 300D0609 2A864886 F70D0101 02050003 81BD0330 81890281
8100AF32 1ED9347E 4473FB F895367F 32058929 9471C960 D35FC0FF 300D0609
9E1E0D66 69800B78 E313C88 1B12650 0FBE01CA B64EF52 0FFE02FD 4D8CEFFA
AEB71024 4E7B42E5 E5C5B4AF E0D619A9 06411555 68FF0236 A7937023 D54D19A8
D6CB0203 10001A3 53305130 0F060355 1D130101 FF040530 030100FF 301F0603
551D2304 18301680 143E4473 FFBF9536 7F32090E 929471C9 6D0535FC FF301D06
03551D0E 04160414 3E4473FB F895367F 32058929 9471C960 D35FC0FF 300D0609
2A864886 F70D0101 05050003 8118010A 083C70BD 5A062FFC 7BBF8F60 49621144
7E0B6C6E 59C85C94 03D11568 7BFA9283 F9C7C5AF AE24A98E 04263221 615D9086
C3AB236D C813EAA4 BAC0C119 4A953E9F FB95B4B3 108C3E08 60A88BB9 51DA07E9
522B3142 43BE17FA 3B15D928 04B00A21 C9D5CB98 DDD06C8E 989EBD23 313DC9BE
65129881 B990074 0AF355C2 7381E5
quit
voice-card 0
dspsfarm
dsp services dspsfarm
!
!
voice service voip
ip address trusted list
ipv4 10.10.0.133
ipv4 10.64.0.127
ipv4 10.70.0.67
ipv4 200.50.67.0 255.255.255.0
ipv4 0.0.0.0 0.0.0.0
ipv4 198.147.226.0
ipv4 198.147.226.0 255.255.255.0
address-hiding
mode border-element
allow-connections sip to sip

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

sip
rel1xx supported "rel100"
min-se 180 session-expires 180
g729 annexb-all
!

voice class codec 1
codec preference 1
g729br8

Codecs used for test: Change order of preference based on codec being tested

For Fax protocol tests – t38 and g711u protocols were tested.
To configure for G711u fax:
use command:

fax protocol pass-through g711ulaw

Enabling CUBE mode

Testing was conducted at tekVisio labs.
codecs preference 2
g711ulaw
codecs preference 3
g729r8

voice translation-rule 1
rule 1 /\+(1..\........\)/ /\1/

voice translation-rule 4
rule 1 /12142425996/ /12142425996/

voice translation-profile addplus
translate called 2

voice translation-profile rmv_one
translate called 3

voice translation-profile strip4
translate called 4

license udi pid C3900-SPE250/K9 sn F0C15391V1LH

hw-module pvdm 0/0

username administrator password 7 105A0C123346085A5C0A

redundancy

translation-
rule 2
Rule 1 6049380601 2629
Rule 2 6049380602 2700
Rule 3 6049380603 2644
Rule 4 6049380604 2604
Rule 5 6049380605 2302

translation-
rule 3
Rule 1 0601 2629
Rule 2 0602 2700

Translation rule for 10 digit (from MetaSwitch) to 4 digit private extensions on Cisco.

Translation rule for 4 digit (from MetaSwitch) to 4 digit private extensions on Cisco.
Rule 3
0603 2644
Rule 4
0604 2604

! translation-rule

4
Rule 1 3001
6049380601
Rule 2 3002
6049380602
Rule 3 3003
6049380603
Rule 4 3004
6049380604

! CUBE WAN side connection to MetaSwitch
CUBE LAN side connection to CUCM

interface GigabitEthernet0/0
description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
ip address 10.70.10.15 255.255.0.0
duplex full
speed 100

! interface GigabitEthernet0/1
no ip address
shutdown
duplex auto
speed auto
!
interface GigabitEthernet0/2
description Connection to Metaswitch
ip address 174.46.0.147
255.255.255.128
duplex full
speed 100 duplex auto
speed auto
!
interface GigabitEthernet0/3
no ip address
shutdown
duplex auto
speed auto
!
ip forward-protocol nd
!
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
ip route 0.0.0.0 0.0.0.0 174.46.0.129
ip route 10.10.0.0 255.255.0.0 192.168.100.1
ip route 10.64.0.0 255.255.0.0 10.70.10.1
ip route 10.70.0.0 255.255.0.0 10.70.10.1
access-list 23 permit 10.10.10.0 0.0.0.7
! nls resp-timeout 1
cpd cr-id 1
!
control-plane
!

voice-port 0/0/0
    station-id name test
    station-id number 16049380602
    caller-id enable
!
voice-port 0/0/1
    station-id name test
    station-id number 16049380604
    caller-id enable
!

mgcp profile default
!
scmp local GigabitEthernet0/0
scmp ccm 10.70.20.2 identifier 1 priority 1 version 6.0
!
scmp ccm group 23
    description tvg MTP
    bind interface GigabitEthernet0/0
    associate ccm 1 priority 1
    associate profile 23 register tvglmtp
!
scmp ccm group 111
    description tvg1 transcoding
    bind interface GigabitEthernet0/0
    associate ccm 1 priority 1
    associate profile 111 register tvgltxcode
!
dial-peer voice 5996 voip
    shutdown
    destination-pattern 21424259..
    session protocol sipv2
    session target ipv4:10.64.1.72:5060
dtmf-relay rtp-nue h245-signal h245-alphanumeric
codec g711ulaw
!
dial-peer voice 9 pots
    description to fax port
destination-pattern 6049380602
    no digit-strip
port 0/0/0
!
dial-peer voice 12 pots
description to fax port
destination-pattern
6049380604
no digit-strip
port 0/0/1
!
dial-peer voice 15 voip
description Trunk to MetaSwitch
destination-pattern 1........
session protocol sipv2
session target
dns:whistler.datcon.co.uk
incoming called-number .%
voice-class codec 1 offer-all
dtmf-relay rtp-nte
!
dial-peer voice 16 voip
description To Cluster4
destination-pattern
604938060[0-9]
translate-outgoing called
2
session protocol sipv2
session target
ipv4:10.70.19.3
incoming called-number .%
voice-class codec 1 offer-all
dtmf-relay rtp-nte
!
dial-peer voice 17 voip
destination-pattern 1866........
session protocol sipv2
session target
dns:whistler.datcon.co.uk
voice-class codec 1 offer-all
dtmf-relay rtp-nte
!
dial-peer voice 18 voip
description to call PBX-PBX
translation-profile outgoing
addplus
destination-pattern 604938060[0-9]
session protocol sipv2
session target
dns:whistler.datcon.co.uk
voice-class codec 1 offer-all
dtmf-relay rtp-nte
!
dial-peer voice 19 voip
description Mapping to private extensions
shutdown
destination-pattern 060[0-9]
translate-outgoing called 3
session protocol sipv2
session target ipv4:10.70.19.3
incoming called-number 3...
voice-class codec 1 offer-all
dtmf-relay rtp-nte
!
dial-peer voice 20 voip
description mapping 4 digit to private
destination-pattern 3...
translate-outgoing called 4
session protocol sipv2
session target dns:whistler.datcon.co.uk
incoming called-number 3...
voice-class codec 1 offer-all
dtmf-relay rtp-nte
!
dial-peer voice 22 voip
description Intl1 calls
destination-pattern 0119180........
session protocol sipv2
session target dns:whistler.datcon.co.uk
incoming called-number .%
voice-class codec 1 offer-all
dtmf-relay rtp-nte
!
dial-peer voice 602 voip
description formidcall neg
huntstop
shutdown
destination-pattern 6049380602
translate-outgoing called 2
session protocol sipv2
session target ipv4:10.70.19.3
incoming called-number .%
dtmf-relay rtp-nte
codec g729br8
!
!
sip-ua
!
!
!
gatekeeper
shutdown
!
!
!
!
banner exec ^CC
% Password expiration warning.
--------------------------------------------------------------------------

Cisco Configuration Professional (Cisco CP) is installed on this device and it provides the default username "cisco" for one-time use. If you have already used the username "cisco" to login to the router and your IOS image supports the "one-time" user option, then this username has already expired. You will not be able to login to the router with this username after you exit this session.
It is strongly suggested that you create a new username with a privilege level of 15 using the following command.

```
username <myuser> privilege 15 secret 0 <mypassword>
```

Replace `<myuser>` and `<mypassword>` with the username and password you want to use.

```
--
Feb 23 20:43:16.176: rtpspi_process_timers: timer expired for type(0x6):
RTP_GLOBAL_INACTIVITY_TIMER
Feb 23 20:43:16.176: rtpspi_handle_rtp_global_timer_expiry:
rtpspi_count_ocb_with_rtttimeout:0 Nothing to do

Feb 23 20:43:16.176: rtpspi_start_rtp_global_timer -------------------------------
--
^C
```

Cisco Configuration Professional (Cisco CP) is installed on this device. This feature requires the one-time use of the username "cisco" with the password "cisco". These default credentials have a privilege level of 15.

**YOU MUST USE CISCO CP OR THE CISCO IOS CLI TO CHANGE THESE PUBLICLY-KNOWN CREDENTIALS**

Here are the Cisco IOS commands.

```
username <myuser> privilege 15 secret 0 <mypassword>
no username cisco
```

Replace `<myuser>` and `<mypassword>` with the username and password you want to use.

**IF YOU DO NOT CHANGE THE PUBLICLY-KNOWN CREDENTIALS, YOU WILL NOT BE ABLE TO LOG INTO THE DEVICE AGAIN AFTER YOU HAVE LOGGED OFF.**

For more information about Cisco CP please follow the instructions in the QUICK START GUIDE for your router or go to http://www.cisco.com/go/ciscocp

```
^C
!
line con 0
    login local
line aux 0
line vty 0 4
    exec-timeout 0 0
    privilege level 15
    logging synchronous
    login local
    transport input telnet ssh
line vty 5 15
    exec-timeout 0 0
    privilege level 15
    logging synchronous
    login local
    transport input telnet ssh
```
! scheduler allocate 20000 1000
end
Configuring the Cisco Unified Communications Manager

CUCM Version
SIP Trunk Profile:

Figure 2 Non-Secure SIP Trunk Profile

![SIP Trunk Security Profile Configuration](image)

**SIP Trunk Security Profile Information**

- **Name**: Non Secure SIP Trunk Profile
- **Description**: Non Secure SIP Trunk Profile authenticated by null String
- **Device Security Mode**: Non Secure
- **Incoming Transport Type**: TCP+UDP
- **Outgoing Transport Type**: TCP
- **Enable Digest Authentication**: [on/off]
- **Nonce Validity Time (minutes)**: 600
- **X.509 Subject Name**: 
- **Incoming Port**: 5060

**Notes**:

- * indicates required item.
- ** If this profile is associated with an EMCC SIP trunk, Accept Out-of-Dialog REFER is enabled regardless of the setting on this page.

Testing was conducted at teMizion labs.
**SIP Profile:**

**Figure 3 SIP profile: Standard SIP Profile - Early Offer (1/2)**

![SIP Profile Configuration](image)

<table>
<thead>
<tr>
<th>Status</th>
<th>Status: Ready</th>
</tr>
</thead>
<tbody>
<tr>
<td>Info</td>
<td>All SIP devices using this profile must be restarted before any changes will take effect.</td>
</tr>
</tbody>
</table>

### SIP Profile Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name*</td>
<td>Standard SIP Profile - early offer</td>
</tr>
<tr>
<td>Description*</td>
<td>Default SIP Profile</td>
</tr>
<tr>
<td>Default MTP Teledphony Event Payload Type*</td>
<td>101</td>
</tr>
<tr>
<td>Resource Priority Namespace List</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Early Offer for G. Clear Calls*</td>
<td>Disabled</td>
</tr>
<tr>
<td>SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*</td>
<td>TIAS and AS</td>
</tr>
<tr>
<td>User-Agent and Server header information*</td>
<td>Send Unified CM Version Information as User-Agent</td>
</tr>
<tr>
<td>Redirect by Application</td>
<td>Yes</td>
</tr>
<tr>
<td>Disable Early Media on 180</td>
<td>Yes</td>
</tr>
<tr>
<td>Outgoing T.38 INVITE include audio online</td>
<td>Yes</td>
</tr>
<tr>
<td>Enable ANAT</td>
<td>Yes</td>
</tr>
<tr>
<td>Require SDP Inactive Exchange for Mid-Call Media Change</td>
<td>Yes</td>
</tr>
<tr>
<td>Use Fully Qualified Domain Name in SIP Requests</td>
<td>Yes</td>
</tr>
<tr>
<td>Allow Presentation Shering using BFCP</td>
<td>Yes</td>
</tr>
</tbody>
</table>

### Parameters used in Phone

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)*</td>
<td>180</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)*</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)*</td>
<td>3600</td>
</tr>
<tr>
<td>Timer T1 (msec)*</td>
<td>590</td>
</tr>
<tr>
<td>Timer T2 (msec)*</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE*</td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INVITE*</td>
<td>10</td>
</tr>
<tr>
<td>Start Media Port*</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port*</td>
<td>32766</td>
</tr>
</tbody>
</table>
Testing was conducted at tekVizion labs.
Trunk to CUBE:

Figure 5 Trunk to CUBE (1/3)

<table>
<thead>
<tr>
<th>Device Information</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Product</td>
<td>SIP</td>
</tr>
<tr>
<td>Trunk Service Type</td>
<td>SIP</td>
</tr>
<tr>
<td>Device Name</td>
<td>CUBE</td>
</tr>
<tr>
<td>Description</td>
<td>Default</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Classification</td>
<td></td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Location</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAA Group</td>
<td></td>
</tr>
<tr>
<td>Tunnelded Protocol</td>
<td></td>
</tr>
<tr>
<td>QSIG Variant</td>
<td></td>
</tr>
<tr>
<td>ASK1 ROSE DID Encoding</td>
<td></td>
</tr>
<tr>
<td>Packet Capture Mode</td>
<td></td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
</tbody>
</table>

Testing was conducted at tekVizion labs.
Figure 6 Trunk to CUBE (2/3)
Figure 7 Trunk to CUBE (3/3)

Configuring Route Patterns

Route patterns are configured as below, prefixed by a 4 to identify that a call has to be routed via the CUBE.
### Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>Cisco UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>Cisco UBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
</tbody>
</table>
Important Information

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